

# A Cross-Layer Scheduling Algorithm for DQCA-based WLAN Systems with Heterogeneous Voice-Data Traffic

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**Abstract**— This paper proposes a Cross-Layer priority-based scheduling algorithm for WLAN systems using the DQCA MAC protocol where heterogeneous voice-data traffic is present. In this algorithm voice users have always the highest priority while among the data users, higher priority is given to the user with higher measured Signal-to-Noise ratio. Therefore, each user transmits with the maximum corresponding available rate. The performance of the proposed algorithm is compared to those without Cross-Layer and with the basic DQCA operation without any service distinction. Simulation results show that this enhanced scheme improves the efficiency of the system in terms of data throughput and voice users' capacity. Furthermore, for a given number of voice users, a higher offered data load is supported.

**Index Terms**— Cross-Layer, heterogeneous traffic, rate-adaptive transmission, voice priority scheduling, WLAN, broadband wireless access, QoS.

## I. INTRODUCTION

The research discipline of wireless communications is one of the areas that is recently undergoing major progress and developments. Layering is the dominating design methodology of wireless communications protocol stacks. They have the traditional Open System Interconnection (OSI) layer-based architecture where layer-independency is the main layer principle. Even though this consideration simplifies protocols' design, it seems to be suboptimal for wireless communication systems. This is due to the fact that the wireless medium is available to multiple users that intent to get access and transmit their information. In addition, wireless systems introduce high error rate, burst errors distribution and time varying link capacity.

On the other hand, several Medium Access Control (MAC) schemes have been proposed for WLANs' Data Link Layer. All these protocols manage the radio resources and try to

provide certain Quality of Service (QoS) level to mobile users by avoiding or at least reducing collisions.

The efficiency, in terms of QoS, of such systems can be optimised when considering some vertical information exchange between layers of the protocol stack. This general concept is known as Cross-Layer optimization [1-2]. Therefore, several physical layer parameters should become available to higher layers and, based on this information, higher protocol layers can adapt their behaviour in order to improve network performance. The Cross-layer concept is more evident at the interface between the Physical and the Data Link Control (MAC) layers. In this case, MAC protocols, radio link control, Radio Resource Management algorithms and routing algorithms can benefit from some degree of awareness of the time varying characteristics of the radio channel.

The interest in using cross-layer design issues in mobile communication systems has been growing recently. Regarding to 802.11-based WLAN systems, some Cross-Layer mechanisms have been proposed in the literature. For example, in [3] the authors propose video streaming over 802.11 using a cross-layer approach, considering the Point Coordination Function. They propose an adaptive cross-layer mechanism for enhancing the efficiency of scalable video transmission by performing tradeoffs between throughput, reliability and delay depending on the channel conditions and application requirement. Bouam and Othman in [4] take profit of the knowledge of the 802.11 physical layer's state contained in SNR to manage MAC layer accesses and higher layer tasks. The authors in [5] propose an enhancement in 802.11 MAC without any change in its current MAC protocol structure. The proposed algorithm, called LA, is based on the received signal strength measured from the received frames. The proposed algorithm selects which rate to use for each particular frame transmission.

In this context, this paper proposes the use of a rate-adaptive MAC protocol with distributed queues in order to achieve the efficient use of the available radio resources. The main idea of rate-adaptive MAC protocols is to send data at

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higher rates when the channel quality is good [6]. A modification of this scheme has been presented in [7], where the authors in an ad-hoc network suggest opportunistically sending more packets when the channel is good. Furthermore, the authors in [8] propose a solution to provide “rate proportional fairness” where the 11Mbps stations receive more bandwidth than the 1Mbps stations in 802.11b scenarios.

The aim of this paper is to provide an efficient resource management by designing a scheduling algorithm based on a Cross-Layer optimization mechanism for a multiservice WLAN system. Two types of services, real time voice and non real time data traffic models are applied and the Distributed Queuing Collision Avoidance (DQCA) protocol is used as MAC protocol over 802.11b standard physical layer [9]. Higher priority is always given to voice users and, among the data users, permission to transmit is given to that with the higher measured Signal to Noise Ratio, which means the one with the better channel conditions. This user transmits a packet using the maximum corresponding rate. The efficiency of the system is measured, in terms of the data mean delay, the data throughput and the capacity of the voice users.

The rest of this paper is structured as follows. Section II analyses the DQCA protocol, while section III presents a priority-based scheduling algorithm for the DQCA. Section IV deals with the analysis of the scheduling algorithm with the use of Cross-Layer design issues. The simulation scenario is presented in Section V. Finally, simulation results are analyzed in Section VI and conclusions are done in Section VII.

## II. DQCA OVERVIEW

DQCA is a distributed high-performance medium access protocol designed for WLAN environments that offers an enhanced performance compared to the 802.11 MAC protocol. It is a reservation scheme that eliminates back-off periods and collisions in data packet transmissions. In addition, it supports a random access mechanism when the traffic conditions are low, thus reducing transmission delays.

The aforementioned protocol has the following features:

- Eliminates back-off periods and collisions in data packet transmissions.

- Performs independently of the number of stations transmitting in the system.

- It does not suffer from instability under all traffic conditions like slotted Aloha, but keeps maximum achieved transmission rate if arrival rate keeps growing.

- It inherently includes a Cross-Layer mechanism that properly manages packet data transmissions.

The main goal of DQCA is to offer access to users in a reserved time interval, confining collisions only to this part of

the frame. In order to achieve this, some short time intervals are reserved for access requests in each time frame, where special Request-To-Send (RTS) packets are sent by the transmitting users. The rest of the time frame is assigned to collision-free data transmission. Furthermore, time slots for ACK and control information (Clear-To-Send, CTS) packets in the downlink are also reserved. Fig. 1 shows the frame structure of the MAC protocol.

This protocol is based on two distributed queues, the Data Transmission Queue (DTQ), and the Collision Resolution Queue (CRQ). The first one is related to data packet transmission scheduling, while the second one is devoted to the collision resolution algorithm. These queues are represented in each user by four integer numbers, denoted by TQ, RQ, pTQ and pRQ. Every user has to update these numbers in each time frame based on feedback information broadcasted by the Access Point (AP) in a downlink packet (DL). TQ is the number of messages waiting for transmission in the DTQ, while RQ is the number of collisions waiting for resolution in the CRQ. Likewise, pTQ is the user position within the DTQ and pRQ is the user position within the CRQ. It is worth noting that TQ’s and RQ’s should have the same values at all users (i.e. they represent distributed queues), whereas pTQ and pRQ represent the positions of each user within the queues, and they may be different for each user.

A new user in the system that has data to transmit should check the state of both distributed queues in order to send an access request or to transmit data. If a user is enabled to send an access request, it randomly selects one of the control slots in the control sub-frame. If more than one request is transmitted in the same control slot, collision occurs and the involved users enter the CRQ and assume a valid pRQ. These users wait for a certain period of time defined by their pRQ value and when they reach the first position of the CRQ they send another RTS. While all collision are being resolved, further access requests by incoming users are prohibited in order to avoid unstable conditions. In a case of a unique access request in a time frame, the requesting user will enter in the DTQ getting a valid pTQ value. It transmits its message when pTQ gets the value 1. When DTQ is empty, an Aloha-like data access transmission takes place to reduce access delay.

It must be emphasized that packet scheduling decisions are not made at the AP but every user is responsible for knowing when to transmit. This is done by calculating the aforementioned four integer values according to the information provided in the downlink packet and some predefined rules. All users increase TQ by the number of the RTS packets that are successfully transmitted, since this number corresponds to the number of users that will enter the DTQ in the next frame. The latter users assume pTQ values that point to the end of the DTQ, sorted by the order in which

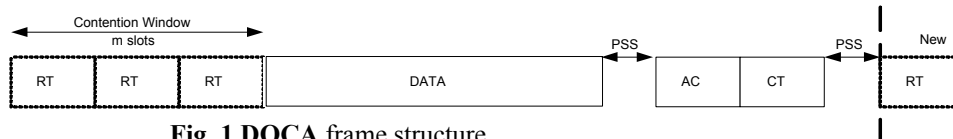


Fig. 1 DQCA frame structure

the RTS have been sent within the Contention Window. In the same way, TQ decreases by 1 when an ACK for data transmission is received. In addition, the users that have a valid pTQ ( $>0$ ) reduce it by 1 and thus advance in the DTQ. In case of collision, the downlink packet indicates the control slot in which it occurred and the number of users involved. The users that have sent an RTS in that slot set their pRQ value and all users update RQ. At this point, only access requests are sent until all users exit CRQ and RQ and pRQ's are reduced for every CTS received.

Due to space constraints, only an overview of the protocol description is included here. Please refer to [10] for a more detailed description.

### III. DQCA PRIORITY BASED SCHEDULING ALGORITHM

DQCA is a fair protocol that is efficient for managing traffic load of any type of service. However, in scenarios of heterogeneous traffic of voice and data, DQCA ignores the specific requirements of each service and treats all users equally, granting access to users according only to their position in the DTQ. The problem is that when the traffic load increases, the QoS requirements of the voice service are not always met. To improve the performance of the system, an alteration in favour of the voice service has been made to the previously described DQCA algorithm. The modified technique will be denoted as DQCA+.

The main idea is to give priority to the voice users that enter the system. In order to achieve that, a third distributed queue, the Voice Transmission Queue (VTQ) has been added to the previous design. This queue is meant to contain all the voice users with packets to transmit, whereas data users are managed by the DTQ. Again, two integers are required to describe the queue, VQ that represents the number of the voice users in the system and pVQ that expresses the position of each user in the VTQ. All users should have a unique value for the VQ and different values for the pVQ.

DQCA+ maintains most features of DQCA, including the reservation access scheme, the collision resolution algorithm and the Aloha-like transmission at low traffic conditions. The differentiation is focused on the data transmission scheduling rules, and only these rules will be described here.

It is assumed that the AP is in position to distinguish between data and voice users' access requests from the type of the RTS packet that is sent. This information is incorporated in the downlink packet and thus is made known to all users. When an RTS from a data user is received, the relevant user derives its pTQ value and all users update TQ. Otherwise, the pVQ and VQ values are calculated. It should be made clear that both DTQ and VTQ are FIFO and the users are sorted by the time of their arrival at the respective queue.

The voice priority is ensured by permitting the DTQ to transmit only when the VTQ is empty. When  $VQ > 0$ , all users are aware that there are voice users in the system with packets ready for transmission. At every frame, the user with  $pVQ = 1$  has permission to transmit. When transmission is completed, all users reduce their VQ and pVQ (if they have a valid one). All data users have to wait until VQ assumes the value zero that means that all voice users have no packets to transmit. At

that point, DTQ resumes control and the data users transmit by order of their pTQ.

### IV. CROSS-LAYER BASED SCHEDULING ALGORITHM

The 802.11b PHY layer supports multi-rate packet transmission, with higher rates being available when the wireless channel quality is good. The proposed MAC protocol acquires the channel state information through a cross-layer dialogue with the PHY layer and implements a rate-adaptive scheme. Its objective is to achieve a more efficient utilisation of the channel bandwidth by granting access to the users that can transmit at the maximum available bit rate ( $R_b$ ), whereas users with lower bit rates wait until the condition of their link improves. This technique will be referred to as DQCA+ with CL, since it maintains the voice priority scheme used in DQCA+ and adds a cross-layer concept.

In the basic DQCA's operation each user sends an RTS to the AP at the minimum transmission rate. The AP calculates the Signal-to-Noise Ratio (SNR) of the link, defines the maximum bit rate at which the user can transmit and includes the  $R_b$  value in the CTS packet. DQCA+ with CL uses this information to perform the rate-adaptive algorithm. The AP, who knows the available bit rates of all the users in the system, rearranges the DTQ by placing users with higher transmission rates to higher positions. If no distinction can be made by means of  $R_b$ , the users are sorted by their arrival at the DTQ. The AP broadcasts a vector with the available  $R_b$  values of all the users by order of their position in the DTQ. Therefore, every user is aware of the condition of all other users and it is in position to derive its own pTQ, that is, its position in the DTQ.

It has to be noted that the cross-layer technique is only applied on the data users. In every other aspect, the protocol resembles to DQCA+ and the voice users have priority over the data users. The VTQ is organised as before and the voice users transmit according to their pVQ, regardless of their channel state.

### V. REFERENCE SCENARIO

The performance of the proposed techniques has been evaluated in an infrastructure WLAN system where a number of data and voice users communicate with an Access Point through a shared radio channel. The physical layer follows the 802.11b specifications and four available rates of 1, 2, 5.5 and 11 Mbps are supported.

#### A. Channel Model

The channel model calculates the probability of each user transmitting at a certain bit rate. The model implements a four-state discrete Markov chain, where each state represents one of the four available bit rates. The idea is based on the fact that although wireless channels are characterized by fast-fading, some correlation exists between the current transmission state and the immediate previous state. This means that, for instance, an 802.11b user able to transmit at 1 Mbps, will probably not be able to transmit at 11 Mbps after a period of time smaller or similar to the coherence time. The Markov chain is represented by a transition matrix  $P$ . This matrix is

used by each user in the system to select the new transmission rate every time the coherence time has elapsed. Without losing generality, the actual matrix  $P$  used in the scenario is:

$$P = \begin{matrix} & \begin{matrix} \text{future} \\ \text{state} \end{matrix} \\ \begin{matrix} \text{current} \\ \text{state} \end{matrix} & \begin{matrix} 1 & 2 & 5.5 & 11 \\ 0.5 & 0.4 & 0.1 & 0 \\ 0.2 & 0.5 & 0.2 & 0.1 \\ 0.1 & 0.1 & 0.5 & 0.3 \\ 0 & 0.2 & 0.3 & 0.5 \end{matrix} \end{matrix}$$

The rows in  $P$  represent the current available transmission rate index. Columns represent the future available transmission rate once the coherence time is elapsed. It should be noted that the selected values for  $P$  are only an example, and they represent a relatively hostile channel in which users are more likely to transmit at the rates of 2 or 5 Mbps. In this situation the Cross-Layer mechanisms can better show their achievable benefits.

It has been assumed that, regardless of the channel conditions, no errors are introduced during transmissions.

## B. Traffic Models

### 1) Data Traffic

Data traffic generation is modelled as Poisson with variable packet sizes that follow an exponential distribution. Their average size is  $10 \cdot L_d$ , with  $L_d$  being the number of bytes transmitted in each frame.

The data service does not accept any loss of packets but can tolerate reasonably large delays. An upper limit of 1 s for data mean delay has been defined.

### 2) Voice Traffic

For the voice users, the ON-OFF model has been adopted, according to which every user can be either active (ON) or silent (OFF). The state diagram of this model is shown in Fig.2. When a voice user enters the system, it is assumed to be at the silent state, during which no packets are generated. At any given time, the user has a probability  $\alpha$  of passing at the active state, where packets of  $L_v = 100$  bytes are generated at a constant bit rate (CBR) of 13 kbps. The probability of the user leaving that state and becoming inactive is  $\beta$ . The mean duration of silent periods  $1/\alpha$  was set to 1,74 s and the mean duration of active periods  $1/\beta$  to 1,41 s.

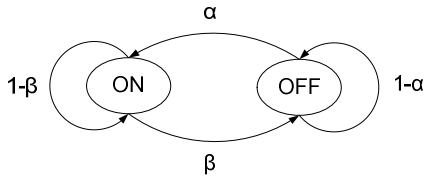


Fig.2 State diagram for ON-OFF model

The voice service can tolerate a maximum delay of 300ms per packet, after which the packet is lost. To provide the desired QoS, only a 1% of the generated voice traffic can be lost.

## C. System Parameters

An heterogeneous system with  $N_d = 20$  data users which generate variable traffic load and a varying number  $N_v$  of voice users has been considered. It has been assumed that all users are moving with 1 m/s speed which corresponds to about 30 ms coherence time at 2.4 GHz. The coherence time defines the frequency at which the AP must update the SNR calculations for every user. The parameters concerning the physical layer were taken from the 802.11b extension of the standard.

The MAC parameters for the three techniques are summarized in Table 1. All control packets (RTS, CTS, ACK) are sent at the minimum rate of 1Mbps in order to ensure reliable transmission. As the RTS packets have no information bits [10], their size is therefore expressed in terms of transmission time. In the case of DQCA+ and DQCA+ with CL, it has been assumed that the RTS packets are of different type (different detection sequence) and hence access requests from data and voice users can be distinguished without the need of extra overhead bits.

Once they get access, data users transmit their messages in consecutive DQCA frames of  $L_d$  information bytes. The same applies to voice users in case of DQCA, with the difference that packets are only 100 bytes and occupy one DQCA frame. DQCA+ and DQCA+ with CL allow the voice users to transmit all the packets in their buffers once they get the turn in transmission, in an effort to reduce unnecessary overhead for such small packets. Therefore, the voice users can send more than one packets with a single RTS.

Table 1. MAC layer parameters

No. of control slots $m$	3
MAC header	34 bytes
RTS	2 $\mu$ s
Data Packet Size $L_d$	1000 bytes
Voice Packet Size $L_v$	100 bytes
Downlink Packet (DL)	13 bytes
DL overhead for DQCA+ with CL	( $2 \times TQ$ ) bits

The downlink packet is transmitted by the AP at the end of each DQCA frame and consists of 2 bytes for the Frame Control (FC) field, 6 bytes for the CTS information, 1 byte for the ACK and 4 bytes for the FCS (Frame Control Sequence).

Finally, the Cross-Layer concept of DQCA+ with CL requires that the  $R_b$ 's of all the users in the DTQ are transmitted within the downlink packet. Since  $R_b$  can assume four values, it can be represented by 2 bits. In addition, the number of users in the DTQ is expressed by the TQ value. Hence, and overhead of ( $2 \times TQ$ ) bits, rounded up to the closest number of bytes, must be added to every downlink packet. This realistic representation of the control traffic ensures that a fair comparison in terms of efficiency can be made with DQCA and DQCA+.

## VI. SIMULATION RESULTS

Simulations were carried out to examine the performance of the three protocols described in sections II-IV.

Given that the number of data users  $N_d$  is constant, the performance of the data service can be described by means of the offered data load, the data throughput and the data mean delay. The data offered load ( $O_d$ ) is the amount of data in bits that arrive at (is generated in) the system at the unit of time. Data throughput ( $T_d$ ) is defined as the ratio of the correctly received data bits to the total simulation time. The data mean delay ( $D_d$ ) is the average waiting time of the data packets from their generation until their complete transmission. On the other hand, the voice service has a constant generation rate and very specific QoS requirements. Therefore it can be fully described by the number of voice users  $N_v$ .

There are two approaches in the evaluation of the performance of the system. One way is to compare the data service statistics ( $O_d$ ,  $T_d$ ,  $D_d$ ) of the three protocols for a given number of voice users  $N_v$ . The second perspective is to maintain the data offered load at a certain level and compare the number of voice users that each protocol can support. The results presented in this section should be viewed by both approaches when possible.

Fig. 3 shows the data throughput versus the offered data load when DQCA, DQCA+ and DQCA+ with CL have been applied. Simulations have been carried out for  $N_v = 5, 10$  and 15 voice users. To provide a complete image of the system's behavior, the data throughput has been plotted regardless of the data QoS requirements.

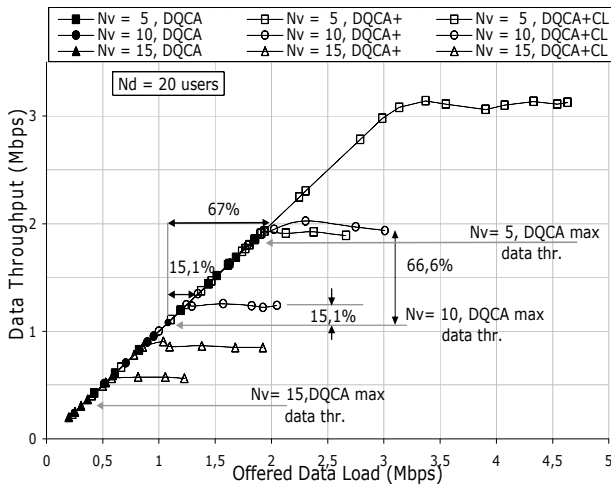


Fig. 3. Data throughput versus offered data load

Ideally, data throughput should be equal to the offered data load since all the data packets that enter the system are to be transmitted at some point. This happens in the case of DQCA where the data throughput is increased linearly with the offered data load, until the voice service suffers from packet losses that exceed 1% and the simulation has been ended. It can be seen that DQCA+ and DQCA+ with CL offer higher data throughput and support more data load for the same number of voice users. However, the shape of the curves is different than

the one corresponding to DQCA, since they increase linearly and when they reach a maximum value of data throughput they stabilize. At the non-linear part of the curves the offered data load increases but the throughput remains constant, meaning that the incoming data packets are not being transmitted but remain stored at the users' buffers. The linear part of the curves expresses the actual enhancement in terms of the offered load and corresponds to mean data delay  $< 1s$ , as demanded by the data QoS.

The improvement introduced by the use of that DQCA+ and DQCA+ with CL with respect to DQCA for  $N_v = 10$  voice users has been marked on Fig. 3 and refers only to the linear parts of the curves. DQCA+ provides an increase of 15.1% in the system throughput. In the case of DQCA+ with CL, by applying the rate-adaptive algorithm, the users transmit at the highest available rate. Therefore, the number of bits sent per frame increases and so does throughput by a factor of 66.6%. Table 2 summarizes the increase in throughput when DQCA+ and DQCA+ CL are used, with respect to the DQCA performance.

Table 2. Data throughput increase with regard to the DQCA performance

Number of Voice Users $N_v$	Increase in data throughput (%)	
	DQCA+	DQCA+ CL
15	43.59	133.07
10	15.12	66.59
5	4.20	60.83

In order to have a more clear perception of the performance of the three techniques, the data mean delay versus the offered data load was measured and plotted in Fig. 4. The trade-off between supported offered data load and data mean delay is very clear. As expected, data packets in DQCA+ and DQCA+ with CL suffer from higher delays but can operate under heavier data conditions. On the other hand, DQCA maintains the data mean delay very low (less than 200 ms) but fails to support larger amount of data load, since the voice service requirements are not met.

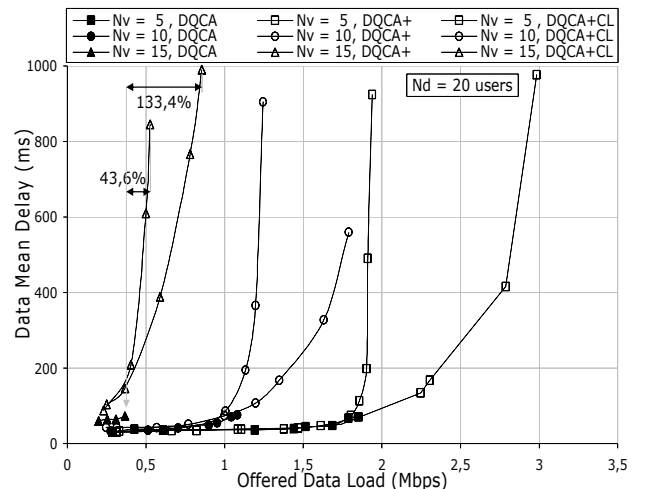


Fig. 4. Data mean delay versus offered data load

For  $N_v = 15$  voice users the improvement in terms of the offered data load has been marked on Fig. 4. DQCA+ gives an amelioration of 43.6% in comparison to DQCA. DQCA+ with CL enhances the amount of the offered data load by 133.4% with an impact on the data mean delay similar to DQCA+. It is true that both techniques cause a clear increase of the data mean delay which can be tolerated by the system since it lies within the margins set by the data service QoS demands.

The last plot of Fig. 5 shows the maximum data throughput for a varying number of voice users. Two types of comparison can be derived. Along the x-axis, comparisons can be made between the data throughput values of the three techniques for the same number of voice users. Along the y-axis, it can be seen the maximum number of voice users that can guarantee a given level of data throughput. For example, for  $N_v = 7$ , DQCA has a data throughput of 1.48 Mbps. DQCA+ increases this value by 10% and DQCA+ with CL gives an enhancement of 76% with a data throughput that reaches 2.6 Mbps. On the other hand, to maintain the data throughput at 2 Mbps, the system can only support 3 voice users with DQCA, 4 with DQCA+ and 9 voice users with DQCA+ with CL.

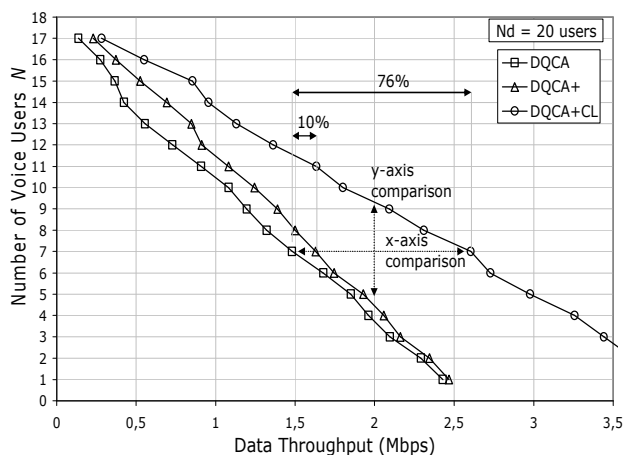


Fig. 5. Maximum throughput per number of voice users

An overall consideration of Fig. 5 shows that DQCA has the lowest performance in terms of the data throughput and the maximum number of voice users. DQCA+ improves this performance, mainly under milder data traffic conditions or when the number of voice users is high. Finally, DQCA+ with CL provides a significant enhancement under all traffic conditions and regardless of the number of voice users.

## VII. CONCLUSIONS

The performance of a WLAN system using DQCA protocol in scenarios of heterogeneous traffic (data and voice) has been examined and two techniques that enhance its performance have been presented.

DQCA+ manages to reduce the loss of voice packets due to large delays. This technique achieves a considerable increase of up to 43.6% of the data throughput for the same number of voice users. This is achieved by increasing the data service

average delays, but could be very useful when this data corresponds to a non real time service.

DQCA+ with CL performs a rate-adaptive strategy that increases even more the data throughput by giving access to users when their channel state allows transmissions at maximum rate. This technique provides an enhancement from 60,8% up to 133% in data throughput. It also manages to increase the total number of voice users that can be supported by the system without reducing the supported data traffic load.

The simulations have shown clearly that in case of heterogeneous traffic, an important advantage can be gained when the two services are treated in a different way, according to their QoS requirements. In addition, the Cross-Layer dialogue enables the design of more efficient schemes and is therefore a promising field of investigation for future wireless communications systems.

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